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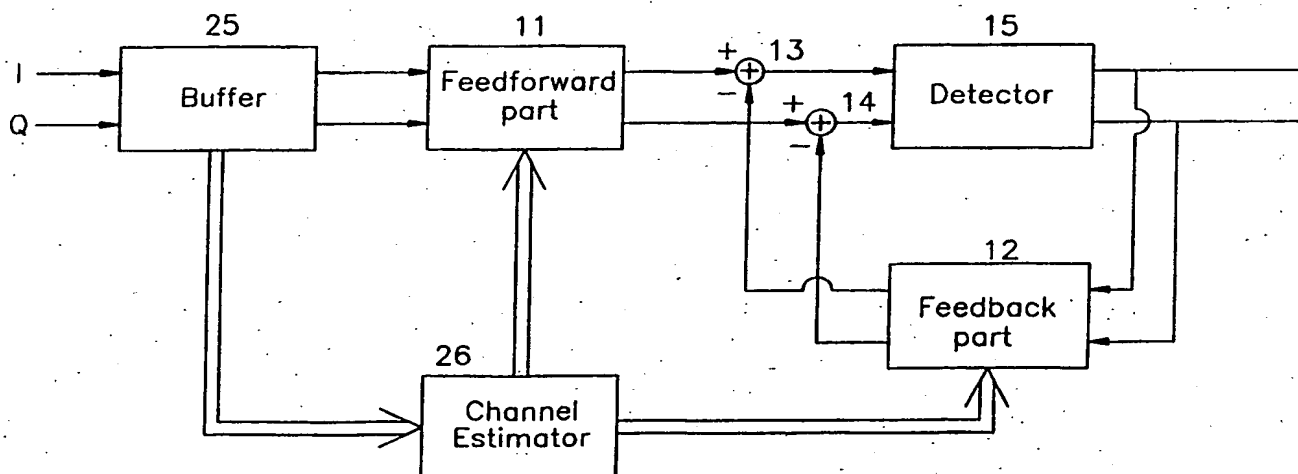
(57) Abstract

In an equalizer of the DFE type (Decision Feedback Equalizer) for equalization in a receiver for electromagnetic signals having passed a transmission channel, a plurality of weight factors or tap coefficients is used in a feedforward part as well as in a feedback part. Values based on an estimate of the impulse response of the channel are selected as a start condition of these coefficients. The estimate values are used in complex conjugated state in the feedforward part, which corresponds to a so-called matched filter, and the autocorrelation of the estimate of the channel impulse response is used in the feedback part. Detection of the data bits can begin at an optional training sequence and proceed from this. The estimate of the impulse response of the channel may also be used for synchronization, and the received signals may be phase or frequency adjusted before detection.

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A method of equalization in a receiver of signals having
passed a transmission channel

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5   The invention concerns a method of equalization for use in
a receiver for electromagnetic signals having passed a
transmission channel. The equalization is of the type
where the received signals are passed to a first series
connection of a plurality of time delay elements which are
10   connected to a common summation point in accordance with a
first set of respective weight factors to produce an out-
put signal. This output signal is passed to a second se-
ries connection of a plurality of time delay elements
which are correspondingly connected to said summation
15   point in accordance with a second set of respective weight
factors.

When communicating over a mobile channel, such as e.g. in
a modern mobile telephone system, the signal between
transmitter and receiver is subjected to various forms of
20   amplitude and phase influences. The influence most inter-
esting in this connection is the one occuring because of
multipath propagation of the transmitted signal. The re-
ceiver receives various contributions which have traversed
various transmission paths, and these contributions inter-
25   fere destructively or constructively depending upon the
frequency. When the mobile unit moves, big fluctuations
will therefore occur in the received power. When digital
signals are transmitted, and when the delay caused by the
30   multipath propagation is greater than the symbol time,
i.e. the time between each information bit, the individual
bits will moreover interfere with each other.

This is called intersymbol interference (ISI). To compen-
35   sate this it is necessary to use an equalizer in the re-
ceiver.

The transmitted data often include a so-called training
sequence, i.e. a bit pattern, which is known in advance by
the receiver. When the received signals are compared with
the expected ones, the momentary impulse response of the
5   channel will be known.

Compensation can be performed in two different ways with
the prior art.

10   In one method no equalization proper of the received sig-
nal takes place. Instead, the method relies on the data
receiver knowing all possible data sequences and being
capable of calculating all possible distorted signals with
its knowledge about the momentary impulse response of the
15   channel and then selecting the one which is the best match
of the signal just received. This type is called Maximum
Likelihood Sequence Estimator (MLSE).

In contrast, the second method relies on equalization of
20   the received signal, i.e. the influence of the transmis-
sion channel on the signal is equalized, following which
the data information is estimated in a simple detector.
Feedback of the estimated data frequently takes place in
the equalizer, and this type is called Decision Feedback
25   Equalizer (DFE). A DFE equalizer is composed of i.a. two
main blocks, viz. a feedforward part and a feedback part.
Each of these comprises a series connection of a plurality
of delay elements whose delay corresponds to the symbol
time, i.e. the time between each information bit so that a
30   data sequence can be shifted through the chain. The re-
ceived data sequence is used as an input signal for the
feedforward part, while the corresponding detected se-
quence is used for the feedback part. The momentary output
signal from each delay element in both the feedforward
35   part and the feedback part is multiplied by a weight fac-
tor or coefficient and is passed to a common summation

point, which is in turn connected to a detector. Generally, these equalizers are adaptive, i.e. they currently adapt to the momentary impulse response of the channel so that the distortion of the channel is equalized. This
5   takes place by current adjustment of the weight factors.

Such equalizers of the DFE type are known from i.a. the European Patent Application 323 870.

10 However, tests with DFE have so far given disappointing results in comparison with MLSE receivers, presumably because they do not have time to converge sufficiently rapidly. On the other hand, the MLSE receivers require much more calculation and are thus more expensive with respect
15 to development as well as production than DFE receivers.

The object of the invention is to provide an equalizer of the DFE type which has a performance of the same order as the one obtainable with MLSE receivers.
20
This object is achieved by selecting a complex conjugated estimate of the impulse response of the channel as start conditions for the first set of weight factors or tap coefficients. An estimate of the impulse response of the
25 transmission channel, expressed by a plurality of filter coefficients, can be calculated in a simple manner by means of the known training sequence, and an optimum solution with respect to noise is obtained by complex conjugating these and then using them as start values for the
30 coefficients in the feedforward part, i.e. the first series connection of a plurality of time delay elements. These coefficients correspond to those used in a so-called matched filter, i.e. a filter whose impulse response is adapted to the transmission channel to provide a maximum
35 signal/noise ratio on the output.

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When additionally, as mentioned in claim 2, generating the
autocorrelation of the estimate of the channel impulse re-
sponse, likewise expressed by a plurality of filter co-
efficients, and using these as start values for the co-
5   efficients in the feedback part, i.e. the second series
connection of a plurality of delay elements, intersymbol
interference originating from symbols i.e. information
bits, detected previously, will be removed.

10   Since the mentioned training sequence will typically be
positioned in the center of a databit sequence, the best
result is obtained by performing the detection from the
center and outwardly, as stated in claim 3, instead of
following the order in which the bits are received.

15

To ensure synchronism with the transmitted signal, the
synchronization time is determined by means of the esti-
mate of the channel impulse response, as appears from
claims 4, 5 and 6.

20

Likewise, it may be necessary to phase or frequency adjust
the received signal according to claim 7 before the actual
detection takes place.

25   In a particular embodiment the method is used in an equa-
lizer adapted for use in receivers for the new common
European digital mobile telephone system called GSM.

The invention will be explained more fully below with re-
30   ference to the drawing, in which

fig. 1 shows an example of the structure of a transmitter/
receiver known per se for a digital mobile telephone sys-
tem,

35

fig. 2 shows a block diagram of a known equalizer of the DFE type,

fig. 3 shows the structure and mode of operation of the feedforward part,

fig. 4 is an example of a simple embodiment of a detector circuit incorporated in the equalizer,

fig. 5 shows the structure of the feedback part,

fig. 6 shows a block diagram of a DFE equalizer provided with a channel estimator,

fig. 7 shows the structure of the channel estimator block,

fig. 8 shows a block diagram of a DFE equalizer with phase of frequency adjustment,

fig. 9 shows an example of the structure of a reconstruction filter,

fig. 10 shows a delay block,

fig. 11 shows a phase adapter unit, and

fig. 12 shows the mode of operation of a phase shifter.

Fig. 1 shows an example known per se of a transmitter/ receiver for a digital mobile telephone system - e.g. the common European digital mobile telephone system called GSM - in which the method of the invention can be used.

A signal is received on an antenna 1 and is passed to a duplexer 2, following which it is amplified and band re- stricted in the high frequence receiver 3. The signal on

its output is complex, the real part being called the I-
component (Inphase) and the imaginary part being called
the Q-component (Quadrature). The complex signal is digi-
tized before being passed further on to the input on the

5   equalizer 4, which will be described more fully below. The
output of the equalizer 4 includes the detected signal
which is passed on for further processing, which can typi-
cally take place in a decoder 5. Further, frequence syn-
thesis 6, control unit 7 as well as the transmitter cir-

10   cuit 8, 9 and 10 are incorporated.

Such a transmitter/receiver is described more fully in
e.g. the International Patent Application PCT/US86/00618.

15   The equalizer 4 may e.g. be of the DFE type, and fig. 2
shows in a block diagram how such an equalizer may be
constructed in a known manner. The DFE equalizer comprises
a buffer 25, a feedforward part 11, a feedback part 12,
the summation points 13, 14 and a detector 15, and these

20   blocks will be described more fully below. The signal from
the buffer is passed to the feedforward part 11, and the
output signal from this is passed to the summation points
13 and 14 (for the real and the imaginary parts, respec-
tively), where the output signal from the feedback part 12

25   is subtracted. The signal is passed from the summation
points 13 and 14 to the detector 15, which may be a com-
parator in its simplest form, and the output signal from
this consists of the detected bits. This signal is passed
partly to the feedback part 12 and partly to subsequent

30   circuits outside the equalizer.

The structure of the feedforward part 11, the detector 15
and the feedback part 12 is described below with reference
to figs. 3, 4 and 5.

35

Fig. 3 shows the structure of the feedforward part. The
input signal X is the received data sequence which is
stored in the buffer 25. The signal is passed through a
plurality of delay elements 16, following which each of
5   the delayed signal values is multiplied by an associated
filter coefficient in the multiplication points 17 and is
summed in the summation points 18. A common summation
point may also be used, as mentioned before. All signals
are complex. The output X' is passed to the summation
10   points 13 and 14 (for the real and the imaginary parts,
respectively).

Fig. 4 shows a simple embodiment of the detector 15, rea-
lized by means of two zero comparators 19, 20 which alter-
15   nately scan the I- and Q-components, respectively, as well
as a switch 21. The output signal is +1 if the scanned
value is greater than or equal to 0, and -1 if it is less
than 0. Correspondingly, the value is purely real if
scanned in the I-channel, and purely imaginary if scanned
20   in the Q-channel.

The detector may also be more sophisticated. The equalizer
of the DFE type per se can just equalize intersymbol
interference from data bits already detected. When using a
25   more sophisticated detector, e.g. a Viterbi detector
allowance can also be made for intersymbol interference
from subsequent data bits.

Fig. 5 shows the structure of the feedback part. The
30   structure corresponds to the feedforward part, incorpo-
rating a plurality of delay elements 22, a plurality of
multiplication points 23 and the summation points 24. The
input signal a is the sequence just detected.

35   If the equalizer is adaptive, the coefficients will be
adjusted currently. This takes place by first selecting a

set of start values or start conditions for each of the complex coefficients. Typically, they are all set to zero, except the real part of one of the coefficients, the so-called main coefficient (main tap) which is set to 1. Then

5   the received training sequence is run through the equalizer, and a comparison with the known training sequence gives an error signal which forms the basis for updating of the coefficients. This procedure can then be repeated until convergence is obtained. However, as mentioned be-

10   fore, it has been found that the equalizer cannot converge sufficiently rapidly in this manner for the desired results to be obtained.

The novelty according to the invention is that the start

15   values of the coefficients for the multiplication points 17 in the feedforward part and the coefficients for the multiplication points 23 in the feedback part are selected in a new manner based upon an estimate of the impulse response of the transmission channel, expressed by a plura-

20   lity of filter coefficients, it being possible to generate such an estimate in a simple manner known per se by means of a channel estimator.

When using these start estimates it is possible to achieve

25   much better results than before, in particular with low signal/noise ratios. Actually, it has surprisingly been found that if the start coefficients are selected in this manner for each received data bit sequence, no additional improvement is achieved by current updating of the coeffi-

30   cients. With low signal/noise ratios (poor receiving conditions) it may even be an advantage not to update the coefficients. This means that the equalizer does not have to be adaptive, which is extremely advantageous with respect to the complixity of the equalizer.

35

In fig. 6 the DFE equalizer from fig. 2 is therefore
equipped with a channel estimator 26. The complex signal
received from the high frequency receiver 3 is first
passed to the buffer 25, which can store the signal for a
5   given period of time, i.e. store a sequence of a given
length. In the GSM system, each received data sequence
contains a training sequence, and as soon as this is
stored in the buffer, the channel estimator 26 can begin
estimating the impulse response of the transmission
10   channel expressed by a plurality of filter coefficients.
This will be described more fully below. In the GSM sys-
tem, the impulse response of the channel comprises con-
tributions from the modulation, transmitter filters, the
physical transmission channel and receiver filters.

15

Fig. 7 shows how the channel estimator 26 may be con-
structed. The input signal X is the received training
sequence which forms part of the received data sequence
stored in the buffer 25. Each element in the received
20   sequence is a complex figure and thus consists of a real
part and an imaginary part. A plurality of delay elements
27 (z^{-1}), a plurality of coefficients 28 (Cn) and a plu-
rality of summation points 29 serve to determine the cross
correlation between the training sequence actually re-
25   ceived and the known training sequence, the latter being
identical with the training sequence transmitted from the
transmitter. The result is a sequence Y of correlation
values which are stored in the buffer 30. Part of this
sequence may be sampled as the estimate (h) of the impulse
30   response of the channel which is used in the feedforward
part 11 according to the invention.

This part is sampled according to a principle where a
window is moved over all correlation values, and the
35   energy of the incorporated correlation values is calcu-
lated for each position. This takes place in that the

sequence Y is passed to a block 31 which calculates the
square norm of the complex values. This gives the sequence
V which is passed to a plurality of delay elements 32, the
number of which corresponds to the length of the window.

5　The sum of the square norms belonging to the window is ob-
tained by means of the summation points 33, which corres-
ponds to the energy of the correlation calculated over the
window length. The block 34 measuring maximum value finds
the window having the greatest content of energy and

10　stores its number, i.e. the time of its beginning or the
reference time, Tref. It is the correlation values in this
window which are used as the estimate of the impulse re-
sponse of the channel.

15　According to an alternative embodiment the method of se-
lecting the estimate of the impulse response of the
channel may be varied. After the window with maximum ener-
gy has been found, it is scanned to find the correlation
value in the window having the greatest square norm, i.e.
maximum amplitude. Then a new window of the correlation is

20　sampled, which is placed symmetrically around the new re-
ference time, and it is then the correlation values in
this which are used as an estimate of the impulse response
of the channel.

25

In both cases the found reference time is passed to the
control unit 7 of the apparatus for synchronization.

The start conditions of the coefficients 17 (feedforward

30　part) and 23 (feedback part) may now be generated with the
reproduced estimate of the impulse response of the channel
as the basis.

The coefficients 17 in the feedforward part are produced

35　according to the invention by complex conjugating (and
time reversing) the values which were sampled in the

channel estimator as an estimate of the impulse response
of the channel.

The coefficients of the feedback part are produced accord-
ing to the invention as part of the impulse response of
the channel folded with the impulse response of the feed-
forward part and thereby as the autocorrelation of the
impulse response of the channel. This is achieved by pas-
sing the estimate values of the window from the correla-
tion buffer through the feedforward part (or an identical
circuit). Only the part of the autocorrelation is used
which corresponds to the rest after the time delay of the
feedforward part has been considered, so as to obtain
correct time synchronism between the two signals in the
summation points 13 and 14.

It is also possible to phase or frequency adjust the re-
ceived signal prior to detection by means of the estimate
of the channel impulse response produced in the channel
estimator 26. Fig. 8 shows how the equalizer from fig. 6
can be equipped with a function for synchronization of
reference frequency. This takes place by detecting and
correcting a phase error. The output signal from the de-
tector 15 is passed to a reconstruction filter 35 which
reconstructs a signal corresponding to the signal either
before or after the feedforward part by means of infor-
mation from the channel estimator 26. The reconstructed
signal is then compared with the actual signal in the
phase adapter unit 36. Since the reconstructed signal is
based on detected values, it will be time delayed, so that
the actual signal must be time delayed correspondingly in
the time delay link 37 before being passed to the phase
adapter unit 36. The phase difference between the two
signals is calculated here. The result of this is passed
to the phase shifter 38 where the received signal is phase
shifted accordingly. Since the system is linear, the phase

shifter 38 may be positioned either immediately before (as shown) or immediately after the feedforward part.

Figs. 9-12 shows the extra blocks which are used in the
5 embodiment from fig. 8 for phase or frequencey correction.

Fig. 9 shows the reconstruction filter which is also composed of delay elements 39, coefficients 40 and summation points 41. As mentioned, the reconstructed signal can correspond to the signal either before or after the feedfor-
10 ward part. If the signal is used before, the impulse response of the reconstruction filter must correspond to the estimate of the impulse response of the channel which is readily accessible in the channel estimator. If the signal is used after the feedforward part, the impulse response
15 of the reconstruction filter must correspond to the impulse response of the channel folded with the impulse response of the feedforward part, which corresponds to the autocorrelation of the estimate, as mentioned before.

20 Fig. 10 shows the delay block consisting of a shift register composed of a plurality of time delay elements 42, which contain time delayed values of the complex input signal. The maximum delay in the block is selected in consideration of the reconstruction filter 35 so as to
25 obtain time synchronism in the phase adapter unit 36. As mentioned, the input signal can be sampled either immediately before or immediately after the feedforward part.

.30 Fig. 11 shows the phase adapter unit determining the phase adaptation on the basis of the phase difference between the output signal from the reconstruction filter and the output signal from the delay block using a phase adaptation constant (μ). The result is a signal which signals
35 whether the phase is to be increased or diminished, and which is passed to the phase shifter.

Fig. 12 shows the phase shifter which accumulates and averages the values in 43 which are received from the phase adapter unit. The output signal is the input signal received from the buffer, phase shifted in accordance with
5   the accumulated phase value. The actual phase shift takes place in the block 44. The sign of the values received from the phase adapter unit is determined by whether the detection takes place from the center of the stored sequence towards its beginning (rearwards) or from the
10   center towards the end.

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Patent Claims:
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1. A method of equalization in a receiver of signals having passed a transmission channel, said signals being passed to a first series connection of a plurality of time delay elements connected to a common summation point in accordance with a first set of respective weight factors to produce an output signal, which is passed to a second series connection of a plurality of time delay elements connected to said summation point in accordance with a second set of respective weight factors, c h a r a c - t e r i z e d by producing an estimate of the impulse response of the channel, complex conjugating said esti- mate, and initiating the first set of weight factors in accordance with a complex conjugated estimate of the im- pulse response of the channel.

2. A method according to claim 1, c h a r a c t e r - i z e d by initiating the second set of weight factors in accordance with the autocorrelation of an estimate of the impulse response of the channel.

3. A method according to claim 1 or 2, wherein the trans- mitted signals are composed of short data bit sequences, and each data bit sequence comprises a subsequence which is known in advance by the receiver and which is so posi- tioned in the data bit sequence in terms of time that it follows and is followed by a plurality of data bits, c h a r a c t e r i z e d by receiving and storing a whole data bit sequence before equalization and detection take place, and initiating equalization and detection in connection with adaptive updating of the weight factors at the known subsequence, said equalization and detection then proceeding away from said known subsequence until the entire sequence has been detected.

4.   A method according to claim 1 or 2, c h a r a c -
t e r i z e d by time synchronizing the equalization to
the received bit stream.

5   5.   A method according to claim 4, c h a r a c t e r -
i z e d by selecting a window, having a suitable length,
of the complex impulse response estimate so that said
window has a maximum content of energy, and using said
window for synchronization.

10

6.   A method according to claim 4 or 5, c h a r a c -
t e r i z e d by synchronizing the equalization by deter-
mining the time corresponding to the maximum amplitude of
the impulse response estimate or a window thereof.

15

7.   A method according to claim 1 or 2, c h a r a c -
t e r i z e d by performing phase or frequency adjustment
of the received signal prior to the actual detection
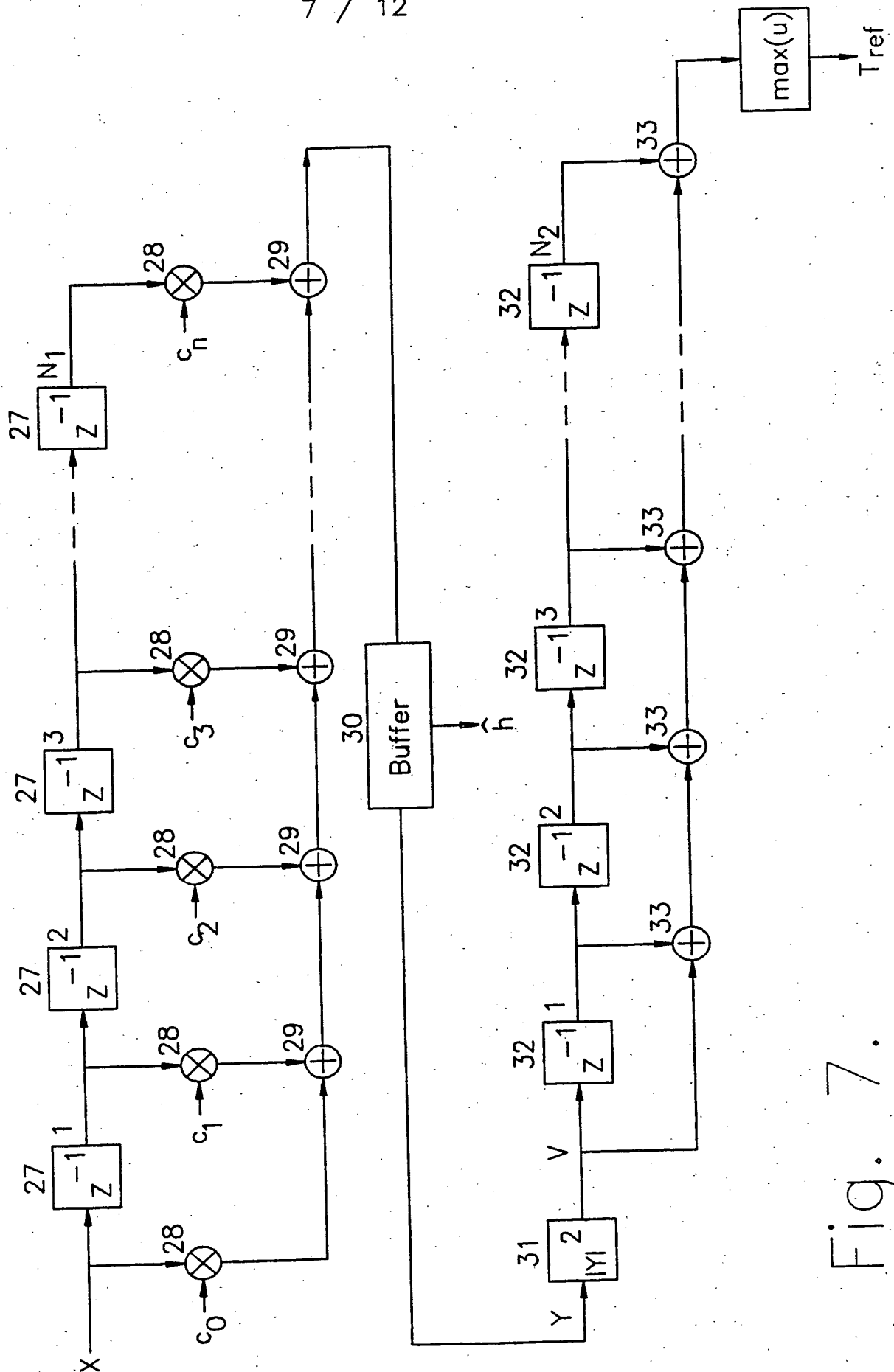thereof.
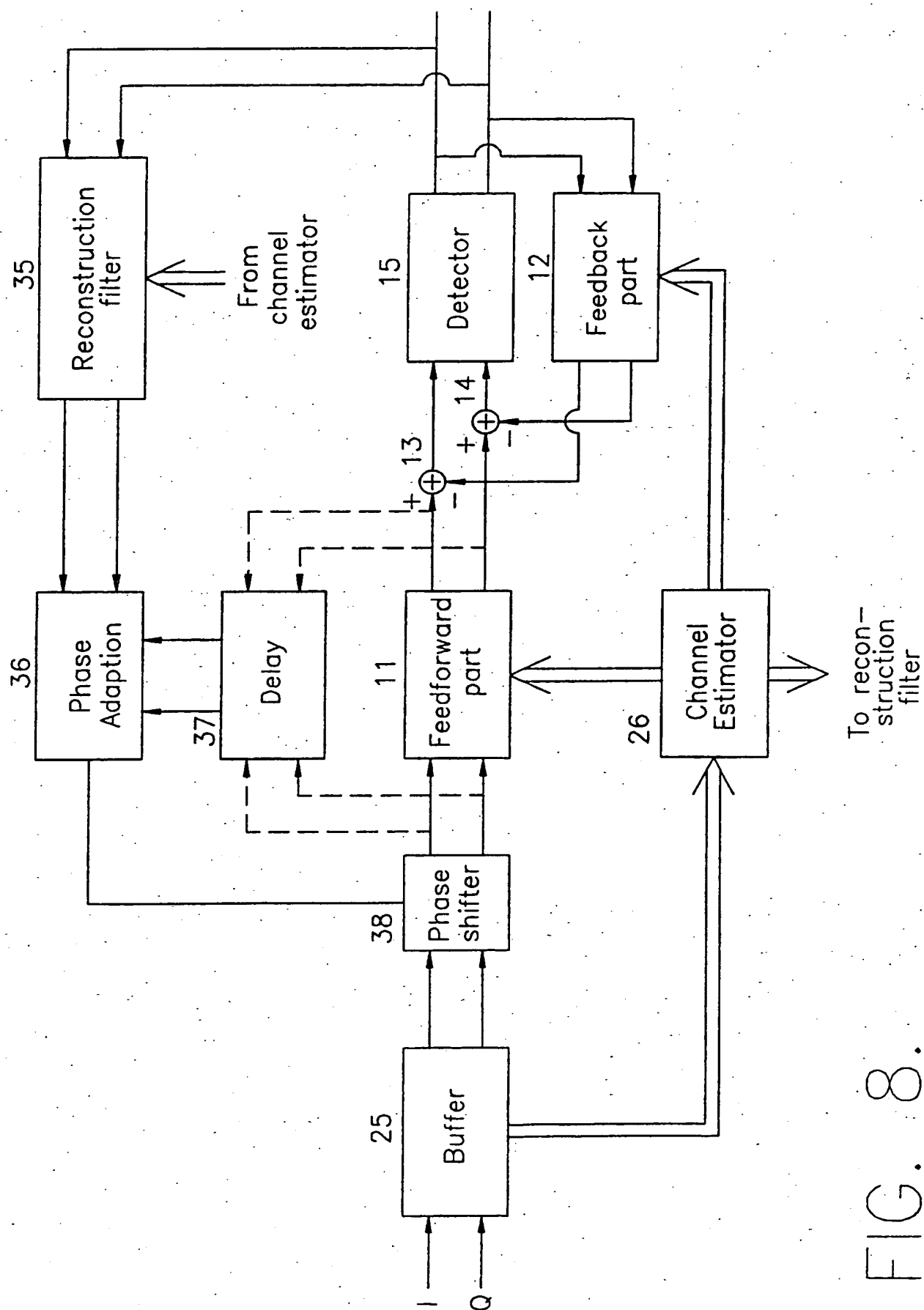
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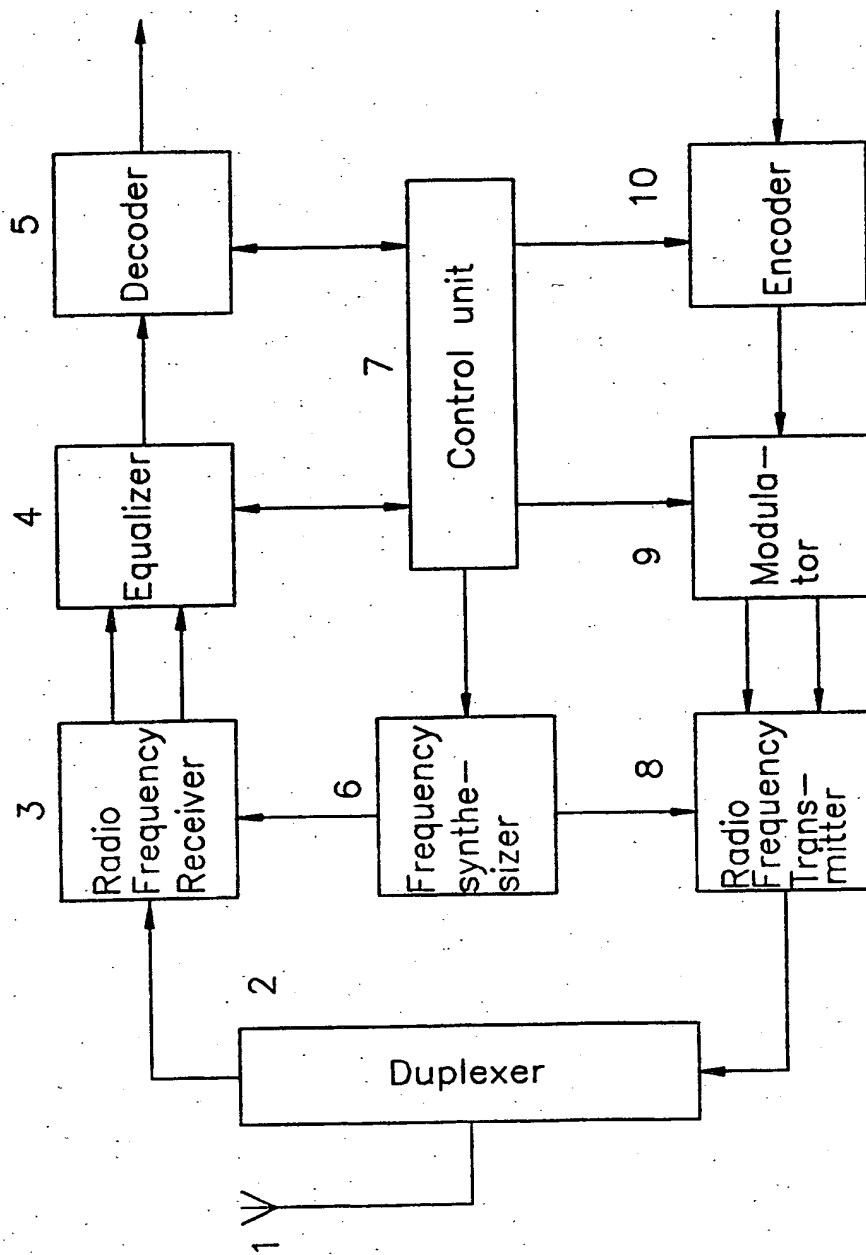
FIG. 1.
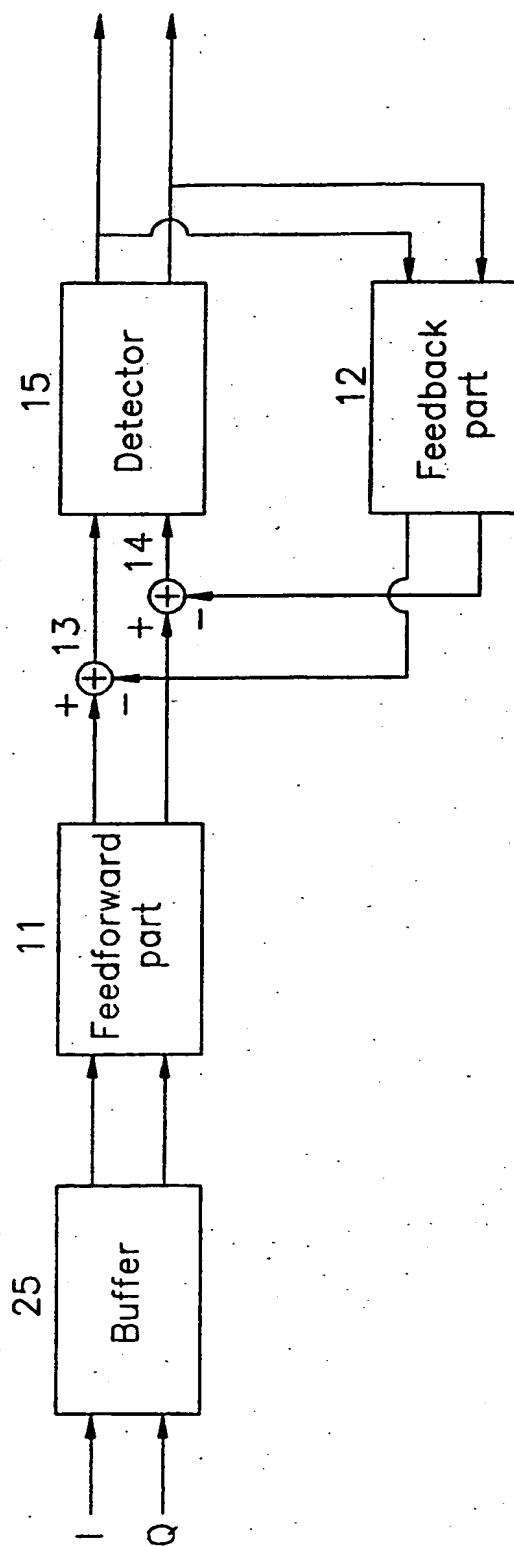
FIG. 2.
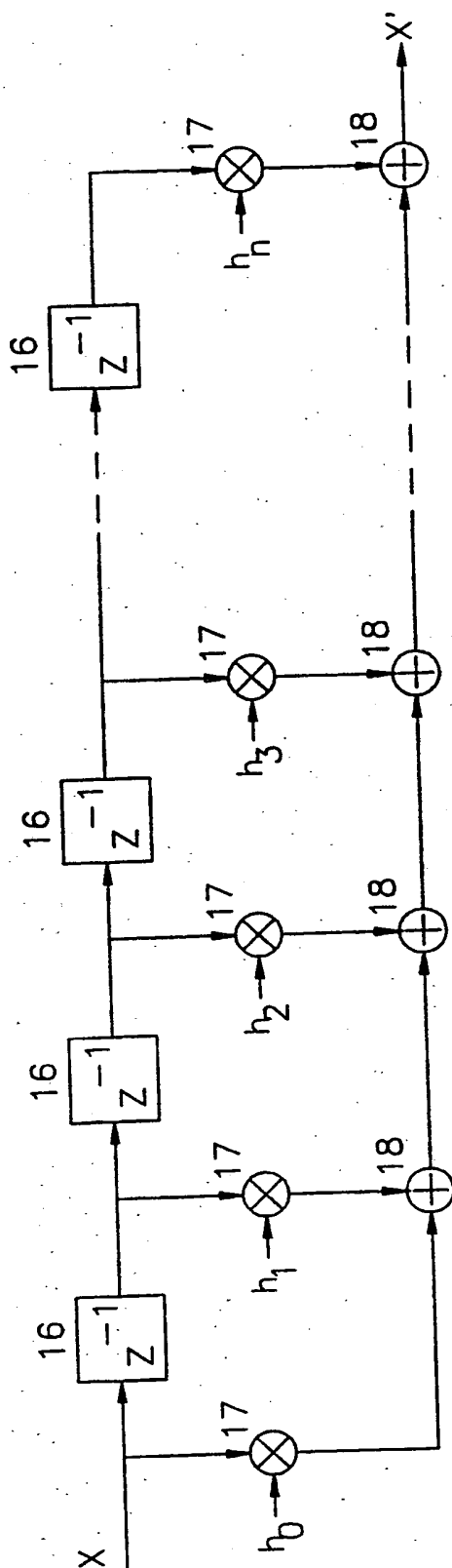
Fig. 3
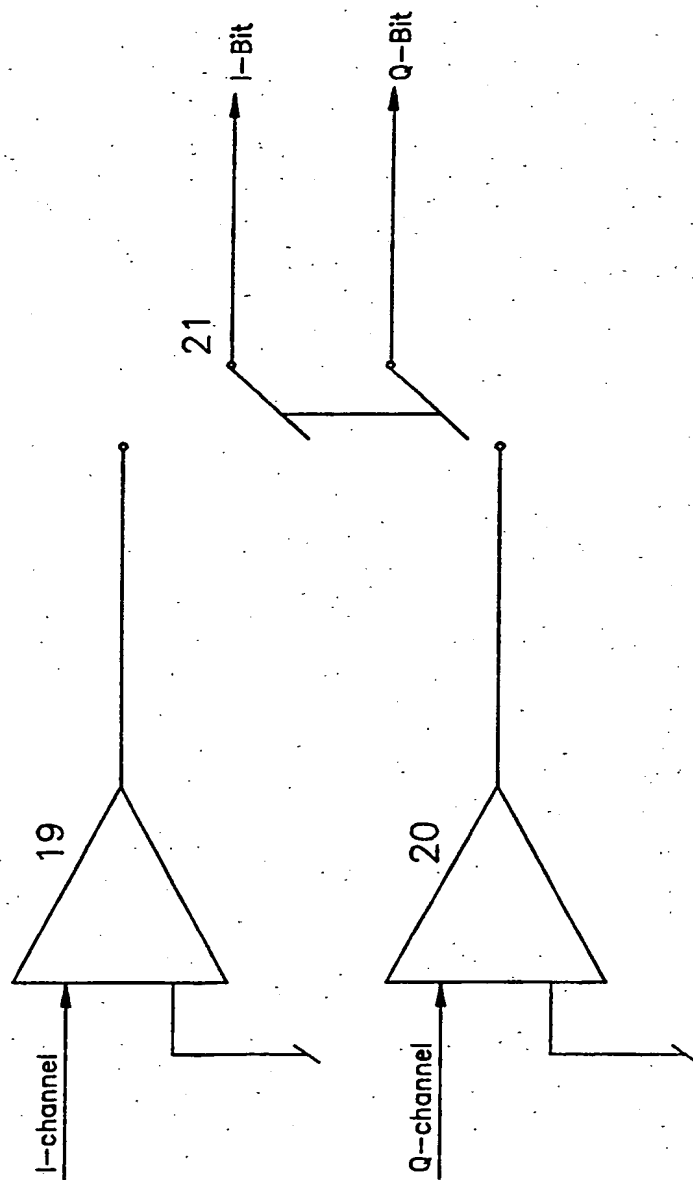
Fig. 4.
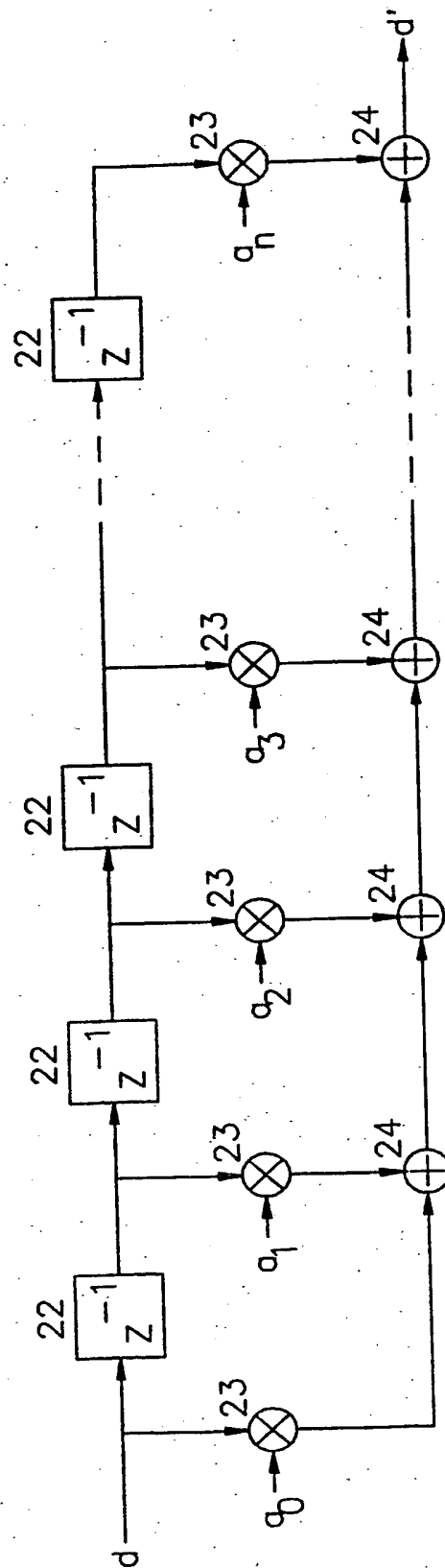
Fig. 5.
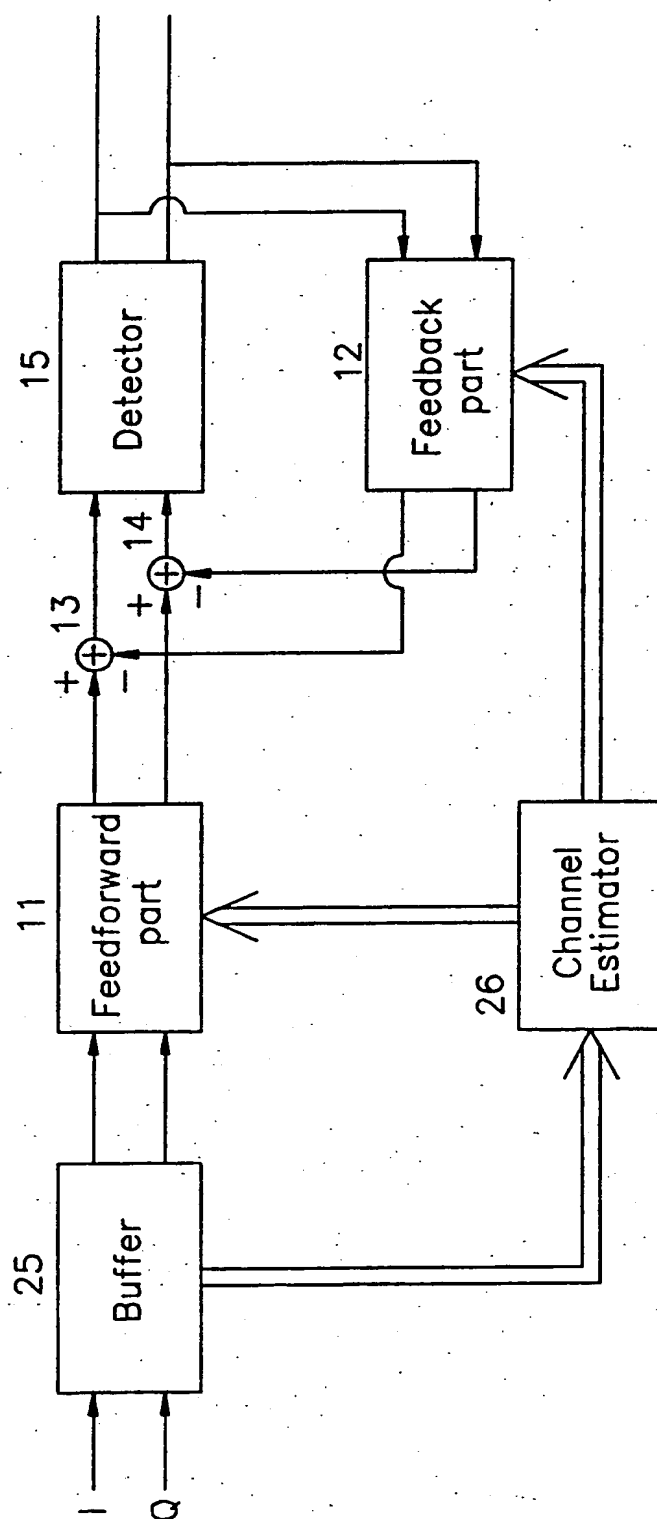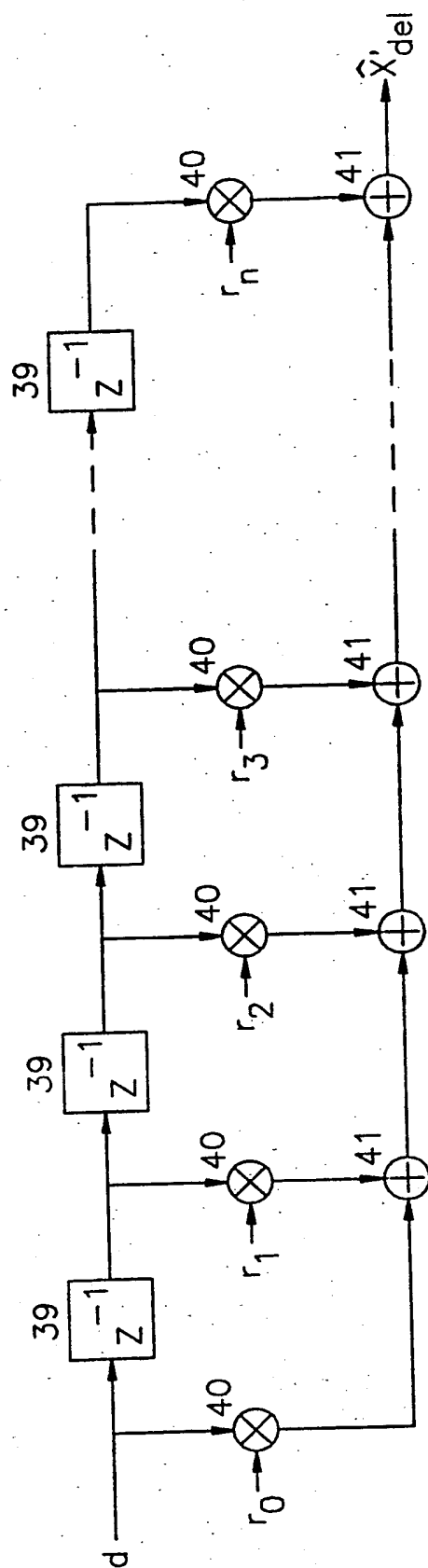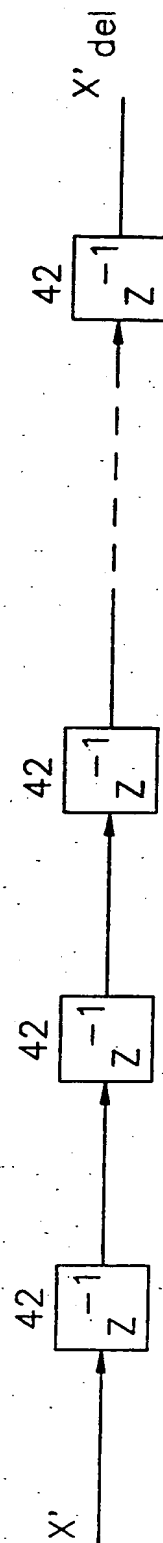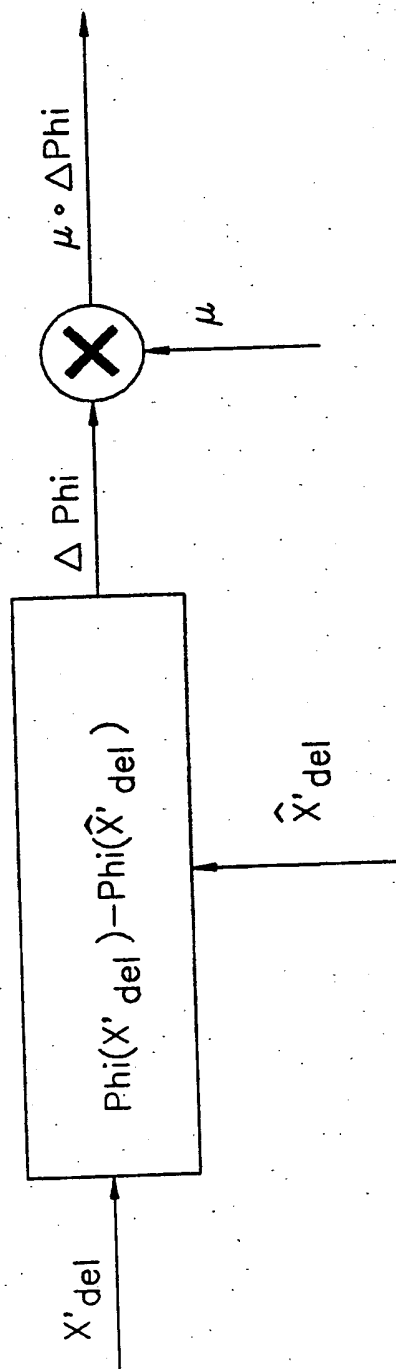
FIG. 6.

Fig. 7.
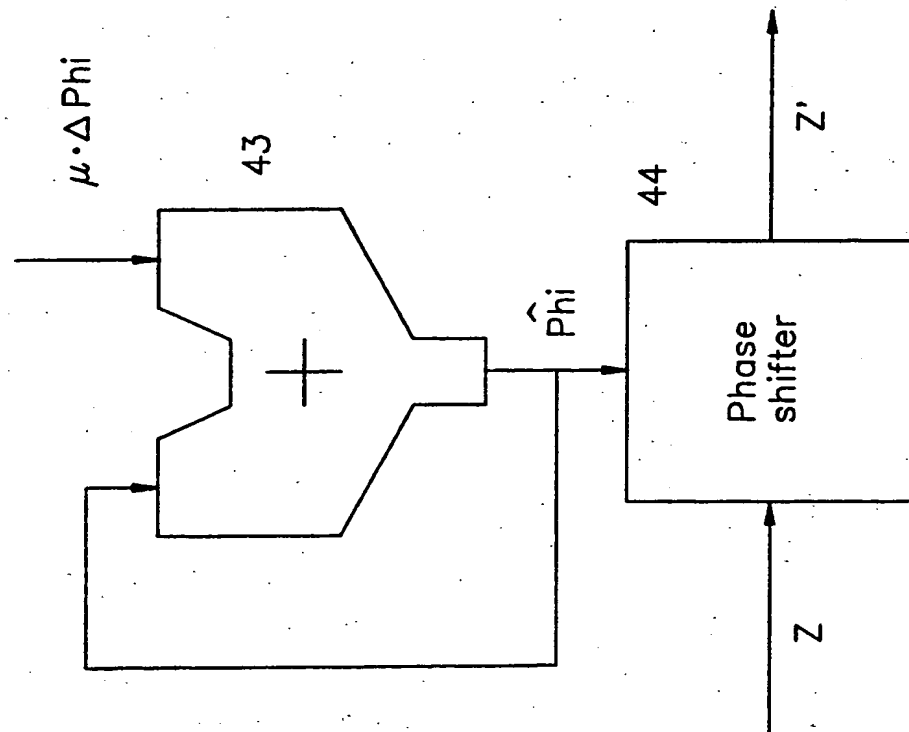
FIG. 8.

Fig. 9.

Fig. 10.

FIG. 11.

FIG. 12.

INTERNATIONAL SEARCH REPORT

International Application No **PCT/DK 91/00114**

I. CLASSIFICATION OF SUBJECT MATTER (if several classification symbols apply, indicate all)[6]

According to International Patent Classification (IPC) or to both National Classification and IPC

IPC5: H 04 B 3/14, 7/005

II. FIELDS SEARCHED

Minimum Documentation Searched[7]

Classification System	Classification Symbols
IPC5	H 04 B

Documentation Searched other than Minimum Documentation
to the Extent that such Documents are Included in Fields Searched[8]

SF,DK,FI,NO classes as above

III. DOCUMENTS CONSIDERED TO BE RELEVANT[9]

Category*	Citation of Document,[11] with indication, where appropriate, of the relevant passages[12]	Relevant to Claim No.[13]
Y	US, A, 4694469 (T. KAKU ET AL) 15 September 1987, see column 13, line 13 - column 16, line 16; figures 20,23	1,2,4,7
Y	US, A, 4701936 (A. CLARK ET AL) 20 October 1987, see column 14, line 25 - line 59; figures 1,5	1,2,4,7
A	US, A, 4430743 (K. WATANABE) 7 February 1984, see column 7, line 14 - line 24; figure 1	1,7
A	US, A, 4571733 (T. KAKU ET AL) 18 February 1986, see column 5, line 15 - column 7, line 21; figure 4	1

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IV. CERTIFICATION

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III. DOCUMENTS CONSIDERED TO BE RELEVANT (CONTINUED FROM THE SECOND SHEET)		
Category *	Citation of Document, with indication, where appropriate, of the relevant passages	Relevant to Claim No
A	US, A, 4441192 (Y. KITA ET AL) 3 April 1984, see column 3, line 23 - column 4, line 65; figure 1	1
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A	GB, A, 2214386 (PHILIPS ELECTRONIC AND ASSOCIATED INDUSTRIES LIMITED) 31 August 1989, see figures 4,5	1,2
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This annex lists the patent family members relating to the patent documents cited in the above-mentioned international search report.
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